



LINK GATE SIP (FIRMWARE VERSION 1.20)



USER GUIDE V1.0



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1. TECHNICAL PARAMETERS

- Dimensions	133 x 233 x 60 mm
- Weight	850 g
- Operating position	various
- Operating condition	temperature: +5°C ÷ +40° C Humidity: 10% ÷ 80% při 30° C
- Power supply	9-15V ss or 8-12Vst, 1.5 A

VoIP

- Ethernet – 10/100Mb with standard BaseT and 100BaseTx, connector RJ45
- SIP connection: P2P or IPPBX SIP server – tested with Cisco Call Manager, Alcatel OMNI PCX, Asterisk, Nexspan, Panasonic...
- 2 VoIP channels (2 IP addresses)
- Codec audio : G711u, G711a, G726, GSM
- VAD (Echo cancellation)
- Protocols: IP, TCP, UDP, http, TELNET, SIP, RTP
- Web server for remote management
- WEB – firmware upgrade
- Remote connection with DTMF dial in
- connection to 2nd operators after time out (adjustable) for direct dialling in
- Caller Id number (CLIP)
- incoming call restriction from GSM network
- outgoing call restriction to GSM network
- Priority connection through either the 1st or the 2nd GSM module (LCR)
- Smart Call back – automatic incoming calls routing up CLIP
- Direct access – assign of IP address to GSM channels
- Echo canceller – switching ON/OFF
- OGM* Option module for recording DISA voice message*
- PIN SIM card protection

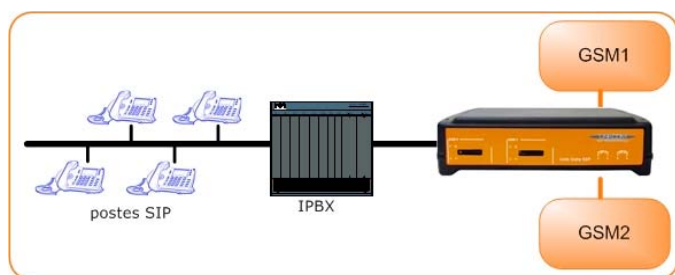
GSM

- GSM 900 (class 4 – 2 W)
- GSM 1800 (class 1 – 1 W)
- Antenna connector SME/SMA, 50Ω
- SIM card: 3/1,8 V
- 2 GSM channels

2. SETTING OPERATIONAL MODE (P2P OR SIP SERVER)

The Link Gate SIP can be configured in 2 different modes: **P2P or SIP Server**.

- **In P2P mode**, the Link Gate SIP behaves as a sample GSM gateway. In case of incoming call on the GSM module, the Link Gate SIP will convert it in SIP and transfer it to the SIP server (IPBX). For an out going call, the Link Gate SIP process a GSM call when SIP call is receives over IP addresses.
- **In SIP Server mode**, the Link Gate SIP acts as a SIP server and allow recording up to 10 IP/SIP phones. It is also possible to register the Link Gate SIP on an existing IPBX.



P2P Mode

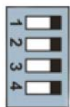


SIP Server Mode



In default mode, the Link Gate SIP is set in SIP server mode, to switch to P2P mode, open the unit and changes the dipswitch position N°3 to ON.

Dipswitch description:



- 1 : IP default Configuration. (OFF = Default position)
- 2 : not used
- 3 : mode, SIP server / P2P (ON = P2P mode / OFF = SIP Server mode)
- 4 : not used

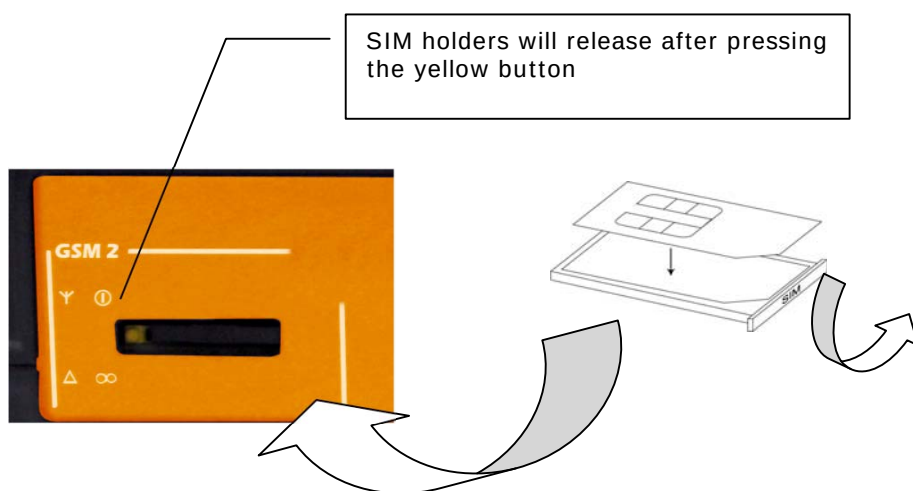
3. PUTTING INTO SERVICE

3.1. MODULE DESCRIPTION

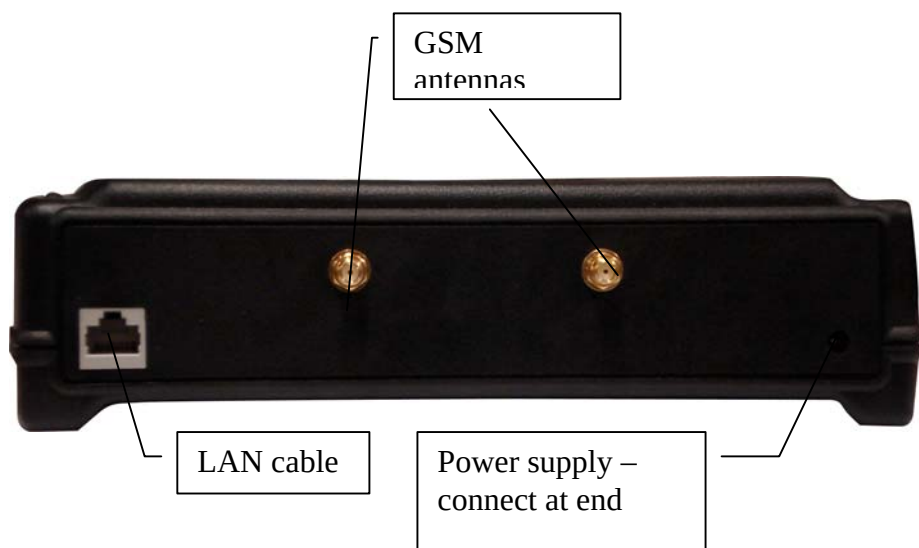
SIM cards position :

Before inserting the SIM card, check the status of PIN code in a mobile phone (starting with PIN or without PIN). **When you want to start with PIN you have firstly to program this PIN to Link Gate SIP. Without this setting the Link Gate SIP will not log to GSM network!**

"We advise not to use PIN Code



SIM card position



Back Panel Vue

Front Panel – LED description

Linkcom France

11 rue du Soleil Levant 92140 CLAMART



Green LED Light: indicates that GSM module is powered.
Red LED: indicates GSM network operation
- Flashing by 2s period = stand by status
- Fast flashing = active connection

Green LED : Indicates the status of Incoming(↓) and outgoing (↑) call via individual IP channels



SIM cards Status
Green LED flashing = module registered to GSM network
Green LED fix light = Call in progress

Red LED indicating channels Error status:
LED is off – stand by mode
LED blink – failure

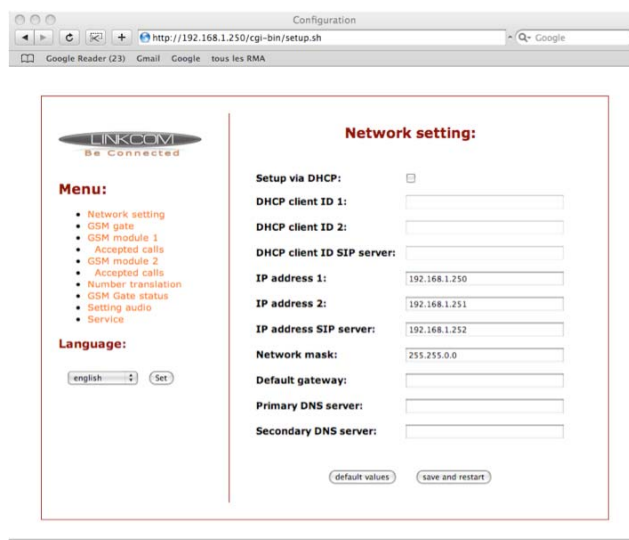
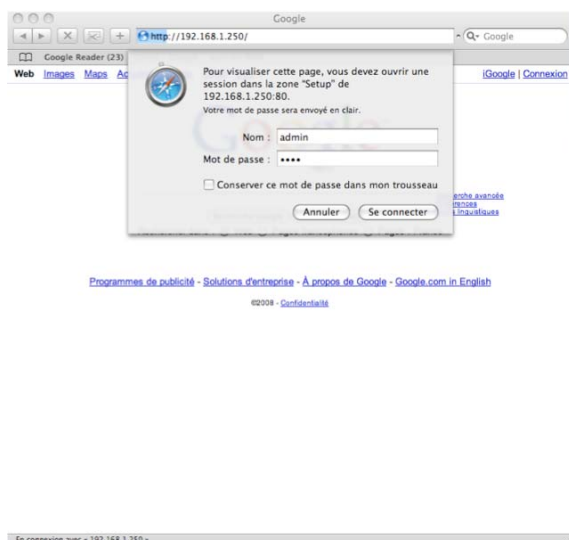
4. NETWORK CONFIGURATION

4.1 - CHOOSING A CONFIGURATION MODE AND LOGIN

By default, the Link Gate SIP is configured as following:

- IP address 1 (IP proxy SIP module GSM 1) = 192.168.1.250 / 255.255.0.0
- IP address 2 (IP proxy SIP module GSM 2) = 192.168.1.251 / 255.255.0.0
- IP SIP Server (SIP Server internal) = 192.168.1.250 / 255.255.0.0

In your web browser enter IP address of Link gate SIP, defaults setting are **192.168.1.250**, **user** is **admin**, and **password** is **1234**.





4.2 - NETWORK SETTINGS: DHCP OR IP SIP MODE

In **Network setting** menu, It is possible to use DHCP automatic setup or enter manually IP addresses. After making all changes click on a **save and restart** button. Restarting is mandatory for those steps

4.3 MANUAL CONFIGURATION IP SIP MODE:

If you prefer to enter manually the IP addresses, make sure to fill out all IP addresses, Network mask, Default Gateway and then all DNS parameters.



4.4 AUTOMATIC CONFIGURATION DHCP MODE:

Tick the Blanc box **Setup via DHCP** and Enter for each IP address ID1 – ID2 the Host name And the **DHCP Client ID SIP Server** name

Enable/disable ethernet settings via DHCP

Identification ID of module 1 for DHCP

Identification ID of module 1 for DHCP

Identification ID of internal SIP Server for DHCP



5- LANGUAGE OPTION:

Select “**Language**” in the left panel menu choose the language then Click **Set** Button.

LINKCOM
Be Connected

Menu:

- Network setting
- GSM gate
- GSM module 1
- Accepted calls
- GSM module 2
- Accepted calls
- GSM Gate status
- SIP parameters
- SIP server
- Setting audio
- Service

Language:

english [v] [Set]

Network setting:

Setup via DHCP: ☐

DHCP client ID 1:

DHCP client ID 2:

DHCP client ID SIP server:

IP address 1:

IP address 2:

IP address SIP server:

Network mask:

Default gateway:

Primary DNS server:

Secondary DNS server:

[default values] [save and restart]

6 – GSM GATE SETTINGS

TO SET UP INCOMING AND OUTGOING CALLS TRUNK ROUTE, REFER TO MENU GSM GATE

6.1 INCOMING CALLS TRUNK ROUTE CONFIGURATION

GSM trunk route settings can be made in the section **GSM Gate module 1** and **GSM Gate module 2**. It is important to set a subscriber extension number, which will be dialled by the module for incoming call. Then it is important to set SIM card PIN, if required. If not, write **0000**. After making changes click on **save changes** button.

LINKCOM
Be Connected

Menu:

- Network setting
- GSM gate
- GSM module 1
- Accepted calls
- GSM module 2
- Accepted calls
- Number translation
- GSM Gate status
- Setting audio
- Service

Language:

english [v] [Set]

GSM Gate - module 1:

PIN:

Subscriber number:

Volume GSM:

Volume VoIP:

Incoming calls: ☒

Outgoing calls: ☒

Call progress tones: ☐

Redirection to GSM 2: ☐

Smart callback: ☐

O:

CLIR: ☐

[default values] [save changes]

SIM card PIN

Set subscriber number



6.2 INCOMING CALLS TRUNK ROUTE CONFIGURATION IN P2P MODE

When the Link gate SIP should be set to P2P mode, it is important to set the translation table for incoming phone calls to match phone subscriber extension's numbers with IP addresses.

After making changes use the **save changes** button.

LINKCOM
Be Connected

Menu:

- Network setting
- GSM gate
- GSM module 1
- Accepted calls
- GSM module 2
- Accepted calls
- Number translation
- GSM Gate status
- Setting audio
- Service

Language:

english

Number translation:

1. Number:	249	IP address:	192.168.1.249
2. Number:		IP address:	
3. Number:		IP address:	
4. Number:		IP address:	
5. Number:		IP address:	
6. Number:		IP address:	
7. Number:		IP address:	
8. Number:		IP address:	
9. Number:		IP address:	
10. Number:		IP address:	
11. Number:		IP address:	
12. Number:		IP address:	

Set a number and IP address, which is set on a desired phone.

6.3. OUTGOING CALL CONFIGURATION:

By default there is no call restriction programmed in the (LCR) table. The Link Gate SIP accepts all SIP calls through its trunk interfaces « **IP 1address** » and « **IP 2address** ». Calls will be routed sequentially over modules GSM 1 et 2.

Set up IPBX External call trunk route parameters (trunk SIP or Outbound proxy sip), SIP phone or soft phone to make calls through the Link Gate SIP. The only parameters that should be configured are IP 1 and 2 addresses and port SIP, 5060 (there is no SIP authentication in mode P2P).

7. SIP SERVER MODE CONFIGURATION

To parameter the Link gate SIP in **SIP server mode**, refer to paragraph 2: “**Setting operational mode**”

In SIP Server mode « **Number translation** » menu is not used

Two new menus are available: « **SIP Parameters** » and « **SIP Server** »

In SIP mode Server the Link Gate SIP can be parameter in 2 ways:

- **SIP server (internal)** : The Link Gate SIP work in this case in stand alone it allow recording up to 10 SIP phone. The Link Gate SIP in this case behaves as an IPBX.
- **SIP server (external)** : The Link Gate SIP is connected and registered to an IPBX as a SIP phone. For each GSM Module one SIP account should be created on the IPBX to allow receiving external calls. Outgoing calls through the GSM channels are made by adding 2 trunks SIP through IP 1 and IP 2 of the Link Gate SIP.

7.1. SIP SERVER MODE (INTERNAL)

Set the Link Gate SIP in MENU **SIP SERVER** tick the Blanc Box to Enable SIP server mode

Give then the Name of the SIP server« **realm** »

Enter the outgoing call prefix to access individually each channel.

You are able then to register up to 10 SIP accounts for your SIP phone.

Complete the number of each SIP phone extension (this number will same as user SIP name in **Password**).

After making changes use button **save and restart**.



SIP Server:

Enable SIP server: ☒ Activate/desactivate internal SIP Server

Server name (realm): SIP Server name

Prefix 1: Prefix for calling through module 1 and 2

Prefix 2:

Number:	Password:
1: 249	123456789
2:	
3:	
4:	
5:	
6:	
7:	
8:	
9:	
10:	

default values save and restart

Table of enabled phones. Set a user name and password, which is set on desired phone

7.2 - SIP SERVER MODE (EXTERNAL)

In « **SIP Server** » Menu make sure that « **Enable SIP server** » is not Tick.

Enter then « **SIP Parameters** » menu and complete IPBX information to register the 2 GSM Modules :

- **Address** : IPBX IP Address
- **Port** : IPBX SIP Port
- **Name** : IPBX extension SIP user name
- **Password** : IPBX extension SIP user name password

Press then « **Save changes** » to save the parameters. The Link Gate SIP will inform you with a message « **Failed Saving** » or « **Successful Saving** »

SIP parameters:

SIP proxy server 1 Address: Name or IP address of external SIP Server for module 1

Port:

Account module 1 Name:

Password:

SIP proxy server 2 Address: Name or IP address of external SIP Server for module 2

Port:

Account module 2 Name:

Password:

default values save changes



7.3 INCOMING CALL SETTING:

In **SIP SERVER** mode Internal or External you will have to select which extension should ring during an Incoming call. fill this Extension N° in “**GSM 1 Module**” and “**GSM 2 Module**”

7.4 OUTGOING CALL SETTING:

In **SIP SERVER** Mode (internal), dial the outgoing prefix number set in “**SIP SERVER**” Menu followed by the phone number.
In our Example GSM 1 Module prefix is 7 and GSM 2 Module prefix is 8.

In **SIP SERVER** Mode (External), create in the IPBX 2 Trunk SIP routes pointing on IP 1 and IP 2 of the Link Gate SIP.

8. DESCRIPTION OF GSM PARAMETERS

8.1 GENERAL GSM PARAMETERS

The screenshot displays the Linkcom web interface for configuring GSM parameters. On the left, a menu lists various settings: Network setting, GSM gate, GSM module 1, GSM module 2, Accepted calls, GSM Gate status, SIP parameters, SIP server, Setting audio, and Service. The 'GSM Gate' option is selected. Below the menu, a language dropdown is set to 'english' with a 'Set' button. The main configuration area, titled 'GSM Gate:', contains several parameters: DISA (set to 0), OGM (checkbox), rec (checkbox), Wait (set to 0), Erase CLIP (set to 0), Modules directly (checkbox), Echo canceller (checked), Add zero before CLIP (checkbox), and Start with module 1 (checkbox). At the bottom of this section are 'default values' and 'save changes' buttons.

DISA: Allow callers to dial-in directly the subscriber number from 1 to 4 digits depending on the numbering schema of the IPBX.

OGM: The Link gate SIP can be provided with OGM optional Board “DISA messages” for individual GSM channel. Allowing recording a DISA welcoming message of 16 seconds, inviting callers to dial the subscriber’s number they want to reach
This message can be recorded through a PC using the special supplied RS 232 interface and the recording software OGMrec.

Rec: The new OGM module doesn't use this parameter.

Wait: when DISA is activated this Time out table allows waiting for caller to dial-in a subscriber number. If caller do not dial in or exceed the Time out figure the call is automatically rooted to the operator.

Erase Clip: Allow erasing first Digits of the Incoming call number.

By default “0” mean no number will be delayed. You can erase the area code of an incoming call ex : +33140831313 will be translated to 140831313.

Modules directly: By activation this feature you disabled LCR (permitted direction). The IP address 1 is automatically assigned to SIM 1 card And IP address 2 is assigned to SIM 2 card. The LCR then can be assign directly in the SIP server.

Echo canceller: This feature restrict ECHO during transfer between digital and analogue telephone systems.

Add zero before CLIP: Adds automatically 0 before each incoming number.

Start with module 1: The Link Gate SIP will start reading LCR in module 1.



8.2. MODULE GSM PARAMETERS

The screenshot shows the Linkcom web interface. On the left, there is a 'Menu' list with options: Network setting, GSM gate, GSM module 1, Accepted calls, GSM module 2, Accepted calls, GSM Gate status, SIP parameters, SIP server, Setting audio, and Service. Below the menu is a 'Language' dropdown set to 'english' and a 'Set' button. The main content area is titled 'GSM Gate - module 1:' and contains the following settings:

PIN:	0000
Subscriber number:	102
Volume GSM:	1
Volume VoIP:	1
Incoming calls:	☒
Outgoing calls:	☒
Call progress tones:	☐
Redirection to GSM 2:	☐
Smart callback:	☐
0:	☐
CLIR:	☐

At the bottom of the settings area are two buttons: 'default values' and 'save changes'.

PIN: Set PIN code of SIM card. If no PIN code is used enter "0000".

Subscriber number: set Subscriber number for Incoming calls

Volume GSM: Adjust from level 1 to 7 the outgoing voice.

Volume ISDN: Adjust from 1 to 4 the Incoming voice.

Incoming calls: Enable incoming calls.

Outgoing calls: Enable outgoing calls.

Call progress tones: Warning tone giving after dialling out a phone number.

Redirection to GSM 2: In case GSM1 channel is busy allow to "overflow" to GSM2 Channel automatically.

Smart callback: Missed or refused outgoing calls are stored in Link Gate SIP with subscribed number.

When called party call back the call is automatically routed to extension which made the call. When connection was successful data are erased from memory.

0: It adds automatically 0 before each outgoing dialled number.

Clir: Switch OFF outgoing CLIP.

8.2. PERMITTED CALLS MENU (LCR)

Permitted call table (LCR) allow for each GSM module to program up to 12 authorised prefixes (1 to 8 digits). If the authorised call table is not filled all type of calls are authorised. We can assign for each of the 12 prefixes a group of Numbers **FROM** xxxxxxxx **To** xxxxxxxx. Ex : **From 0 to 9**.

You can select the for each GSM **Permitted table** the Operator prefixes corresponding to the SIM card inserted,

Ex : GSM 1 Module (T-Mobil) GSM 2 Module (ORANGE) prefixes corresponding to T-Mobile can be inserted in GSM1 Permitted table and for Orange in GSM2 Permitted table in this way you can benefit of cheaper calls.





Be Connected

Menu:

- Network setting
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- GSM module 2
- Accepted calls
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- SIP parameters
- SIP server
- Setting audio
- Service

Language:

english

Permitted calls - module 1:

0.	from:		to:	
1.	from:		to:	
2.	from:		to:	
3.	from:		to:	
4.	from:		to:	
5.	from:		to:	
6.	from:		to:	
7.	from:		to:	
8.	from:		to:	
9.	from:		to:	
10.	from:		to:	
11.	from:		to:	

8.3. GSM GATEWAY STATUS

The **GSM Gate status** provides you with all technical information's corresponding to SIM card in GSM Module 1 and GSM module 2.

Frequency Channel: GSM network channel number - GSM frequency connected to BTS.
Channel 0 - 124 are used for GSM 900 and channel 512 - 885 for GSM 1800.

Signal Strength, BCCH strength: GSM Signal force (Is related with Antenna position)

- o -113 to -99 dBm: very bad signal - it is impossible to use any GSM network services.
- o -98 to -83 dBm: bad signal - you can send SMS messages. Since level -87dBm you can establish voice connection
- o -82 to -71 dBm: good signal - you can send SMS messages and make calls. Data transmission CSD is not reliable
- o -70 to -51 dBm: very good signal - you can use all services of GSM network without any restriction.

Country Code, Network Code, Area Code: Country Code N°, Provider Network N°, where SIM card is logged

Cell ID: BTS number, where GSM module is registered.

IMSI: International Mobile Subscriber Identity

GSM Firmware version: SW version for GSM part.



Be Connected

Menu:

- Network setting
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- Accepted calls
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- Service

Language:

english

GSM Gate status:

	Module 1	Module 2
Frequency channel:	799	119
Signal strength:	16	53
BCCH strength:	-94	-57
Country code:	208	208
Network code:	20	10
Area code:	0006	151A
Cell ID:	7A86	C065
IMSI:	208209850442644	208101169611328
GSM Firmware version:	VER3.B	



8.4. SETTING AUDIO



Be Connected

Menu:

- Network setting
- GSM gate
- GSM module 1
- Accepted calls
- GSM module 2
- Accepted calls
- GSM Gate status
- SIP parameters
- SIP server
- Setting audio
- Service

Language:

english

Setting audio:

Priority 1:

Priority 2:

Priority 3:

Priority 4:

Choose the audio codecs priority for SIP calls. There codecs are available:

- G711u
- G711a
- G726
- GSM

8.5. SERVICE



Be Connected

Menu:

- Network setting
- GSM gate
- GSM module 1
- Accepted calls
- GSM module 2
- Accepted calls
- GSM Gate status
- SIP parameters
- SIP server
- Setting audio
- Service

Language:

english

Admin services:

VoIP version: 1.20

Firmware upgrade:

Upload language:

Service password:

☐ Retype password:

Firmware version

Event log (enhanced or basic log)

Download event log

Firmware upgrade

Language upgrade

Admin password change